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EXAMINER	
MAIS, MARK A	

ART UNIT	PAPER NUMBER
2467	

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Please find below and/or attached an Office communication concerning this application or proceeding.

The time period for reply, if any, is set in the attached communication.

Office Action Summary	Application No. 10/609,117	Applicant(s) TAO, XIN	
	Examiner MARK A. MAIS	Art Unit 2467	

-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --

Period for Reply

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE 3 MONTH(S) OR THIRTY (30) DAYS, WHICHEVER IS LONGER, FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133). Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

Status

1)☒ Responsive to communication(s) filed on 20 August 2009.

2a)☐ This action is **FINAL**. 2b)☒ This action is non-final.

3)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

Disposition of Claims

4)☒ Claim(s) 1,4-6 and 9-20 is/are pending in the application.

4a) Of the above claim(s) _____ is/are withdrawn from consideration.

5)☐ Claim(s) _____ is/are allowed.

6)☒ Claim(s) 1,4-6 and 9-20 is/are rejected.

7)☐ Claim(s) _____ is/are objected to.

8)☐ Claim(s) _____ are subject to restriction and/or election requirement.

Application Papers

9)☐ The specification is objected to by the Examiner.

10)☐ The drawing(s) filed on _____ is/are: a)☐ accepted or b)☐ objected to by the Examiner.

Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).

Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).

11)☐ The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

Priority under 35 U.S.C. § 119

12)☐ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).

a)☐ All b)☐ Some * c)☐ None of:

1.☐ Certified copies of the priority documents have been received.

2.☐ Certified copies of the priority documents have been received in Application No. _____.

3.☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

* See the attached detailed Office action for a list of the certified copies not received.

Attachment(s)

1) ☒ Notice of References Cited (PTO-892)
2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)
3) ☐ Information Disclosure Statement(s) (PTO/SB/08) Paper No(s)/Mail Date _____.
4) ☐ Interview Summary (PTO-413) Paper No(s)/Mail Date. _____ .
5) ☐ Notice of Informal Patent Application
6) ☐ Other: _____.

DETAILED ACTION

Claim Rejections - 35 USC § 102

1. The following is a quotation of the appropriate paragraphs of 35 U.S.C. 102 that form the basis for the rejections under this section made in this Office action:

> A person shall be entitled to a patent unless –
>
> (b) the invention was patented or described in a printed publication in this or a foreign country or in public use or on sale in this country, more than one year prior to the date of application for patent in the United States.
>
> (e) the invention was described in (1) an application for patent, published under section 122(b), by another filed in the United States before the invention by the applicant for patent or (2) a patent granted on an application for patent by another filed in the United States before the invention by the applicant for patent, except that an international application filed under the treaty defined in section 351(a) shall have the effects for purposes of this subsection of an application filed in the United States only if the international application designated the United States and was published under Article 21(2) of such treaty in the English language.

2. Claims 1, 4-6, and 9-20 are rejected under 35 U.S.C. 102(e) as being anticipated by Cave et al. (USP 6,996,094).

3. With regard to claim 1, Cave et al. discloses a method for processing calls in a voice over packet system, the system including a call controller having control modules **[Fig. 4c, VRU 800; call control server (CCS) 802, voice media server (VMS) 804, and application server 803]**, a plurality of media gateways **[Fig. 4c, originating gateway 810 and terminating gateway 812]**, an ingress channel **[Fig. 4c, PSTN connection between phone 814 and originating gateway 810]**, an egress channel **[Fig. 4c, PSTN connection between phone 832 and terminating gateway 812]** and a core packet network **[Fig. 4c, IP network 806]**, the method comprising:

receiving a call having call content, originating information, and terminating information on the ingress channel **[Fig. 4a, VRU 800 receives call (i.e., call content) from phone 814 (PSTN portion between phone 814 and originating gateway 810 is the ingress channel) through originating gateway 810 (caller ID identifies user—originating information), user inputs desired PSTN number of the desired party (terminating information), col. 14, lines 7-60; the call is interpreted as the call to the desired party—thus, only instantiated until after receiving the desired party's PSTN number]**;

establishing an originating half call context for the call based on the originating information **[Fig. 4b, PSTN network routes the call to originating gateway 810 (col. 14, lines 7-9); an RTP/RTCP session established between originating gateway 810 and VRU 800 in order to get the terminating information, col. 14, lines 33-34; Cave et al. also discloses using only control signaling between the gateways and the VRU and connecting the media stream to the destination (i.e., directly connecting the half calls between originating gateway 810 and terminating gateway 812), col. 21, lines 43-48]** the originating half call context having terminating points within one of the plurality of media gateways **[Figs 4b, established at originating gateway 810, col. 14, lines 7-9; the H.323 calls (i.e., terminating points) are not modified, the RTP streams are merely moved, col. 15, lines 45-46]**;

controlling the originating half call context for the call by a first control module of the call controller **[Fig. 4, CCS 802 and application server 803 control the application for incoming requests for 800 calling card service (interpreted as an "originating module"), col. 14, lines 35-40]**;

instructing a second control module of the call controller to establish a terminating half call context for the call **[Fig. 4b, application server 803 then instructs CCS 802 to establish a connection between phone 832 and VMU 800 via terminating gateway 812 (interpreted as a "terminating module"), col. 14, line 61 to col. 15, line 10]**;

establishing the terminating half call context for the call based on the terminating information **[Fig. 4b, a connection is established between phone 832 and terminating gateway 812, col. 15, line 11-23]** the terminating half call context having terminating points within a second one of the plurality of media gateways **[Figs 4c, established at terminating gateway 812, col. 15, lines 13-14; the H.323 calls (i.e., terminating points) are not modified, the RTP streams are merely moved, col. 15, lines 45-46]**;

controlling the terminating half call context for the call by the second module **[Fig. 4c, application server 803 (i.e., using the "terminating module") monitors the call and identifies the called party, col. 15, lines 15-20]**;

transmitting the call content from the originating context to the terminating context based on the controlling of each call context by the first and second control modules **[Fig. 4c, (while keeping open the connections opened by the "originating module" and the "terminating module"), application server 803 directs CCS 802 to transmit RTP streams (call content) directly between originating gateway 810 and terminating gateway 812, col. 15, line 24-46; the H.323 calls (i.e., terminating points) are not modified, the RTP streams are merely moved, col. 15, lines 45-46]**; and,

transmitting the call content out of the system on the egress channel **[Fig. 4c, PSTN connection between terminating gateway 812 and phone 832 is the egress channel]**.

4. With regard to claim 4, Cave et al. discloses that the call content on the ingress channel is in one of time-division multiplexed (TDM) format and packet format **[PSTN, col. 14, lines 7-9]**.

5. With regard to claim 5, Cave et al. discloses that the call content on the egress channel is in one of time-division multiplexed (TDM) format and packet format **[PSTN, col. 15, lines 14-15]**.

6. With regard to claim 6, Cave et al. discloses that the call content is in packet format during the transmitting from the originating call context to the terminating call context **[Fig. 4c, RTP streams between originating gateway 810 and terminating gateway 812 over IP network 806, col. 15, lines 40-43]**.

7. With regard to claim 9, Cave et al. discloses an apparatus for processing calls in a voice over packet system **[Fig. 4c, VRU 800; call control server (CCS) 802, voice media server (VMS) 804, and application server 803]**, the apparatus comprising:

means for receiving a call having call content **[Fig. 4a, PSTN portion between phone 814 and gateway 810 is the ingress channel]**;

means for establishing an originating half call context for the call **[Fig. 4a, at originating gateway 810; Fig. 4b, PSTN network routes the call to originating gateway 810 (col. 14, lines 7-9)]**;

means for controlling the originating half call context for the call **[Fig. 4b, an RTP/RTCP session established between originating gateway 810 and VRU 800 in order to**

get the terminating information, col. 14, lines 33-34; Cave et al. also discloses using only control signaling between the gateways and the VRU and connecting the media stream to the destination (i.e., directly connecting the half calls between originating gateway 810 and terminating gateway 812), col. 21, lines 43-48; the call is interpreted as the call to the desired party—thus, only instantiated until after receiving the desired party's PSTN number; CCS 802 and application server 803 control the application for incoming requests for 800 calling card service (interpreted as an "originating module"), col. 14, lines 35-40];

means for establishing a terminating half call context for the call **[Fig. 4c, at terminating gateway 812]**;

means for controlling the terminating half call context for the call **[Fig. 4b, application server 803 instructs CCS 802 to establish a connection between phone 832 and VMU 800 via terminating gateway 812 (interpreted as a "terminating module"), col. 14, line 61 to col. 15, line 10; then, a connection is established between phone 832 and terminating gateway 812, col. 15, line 11-23; Fig. 4c, application server 803 (i.e., using the "terminating module") monitors the call and identifies the called party, col. 15, lines 15-20]**;

means for transmitting the call content from the originating half call context to the terminating half call context based on the means for controlling **[Fig. 4c, via originating gateway 810; (while keeping open the connections opened by the "originating module" and the "terminating module"), application server 803 directs CCS 802 to transmit RTP streams (call content) directly between originating gateway 810 and terminating gateway 812, col. 15, line 24-46; the H.323 calls (i.e., terminating points) are not modified, the RTP streams are merely moved, col. 15, lines 45-46]**; and,

means for transmitting the call content out of the system on the egress channel **[Fig. 4c, terminating gateway 812; the PSTN connection between terminating gateway 812 and phone 832 is the egress channel]**.

8. With regard to claim 10, Cave et al. discloses means for establishing an originating half call context is a media gateway **[Fig. 4a, at originating gateway 810]**.

9. With regard to claim 11, Cave et al. discloses means for controlling the originating half call context is the call controller **[CCS 802 and application server 803 control the application for incoming requests for 800 calling card service (interpreted as an "originating module"), col. 14, lines 35-40]**.

10. With regard to claim 12, Cave et al. discloses means for establishing the terminating half call context is a media gateway **[Fig. 4c, terminating gateway 812]**.

11. With regard to claim 13, Cave et al. discloses that the originating half call context resides in a media gateway **[Fig. 4a, at originating gateway 810; Fig. 4b, PSTN network routes the call to originating gateway 810 (col. 14, lines 7-9); the H.323 calls (i.e., terminating points) are not modified, the RTP streams are merely moved, col. 15, lines 45-46]**.

12. With regard to claim 14, Cave et al. discloses that the terminating half call context resides in a media gateway **[Fig. 4c, at terminating gateway 812; a connection is established between**

phone 832 and terminating gateway 812, col. 15, line 11-23; the H.323 calls (i.e., terminating points) are not modified, the RTP streams are merely moved, col. 15, lines 45-46].

13. With regard to claim 15, Cave et al. discloses that the means for transmitting the call content from the originating context to the terminating context is a media gateway **[Fig. 4c, via originating gateway 810; (while keeping open the connections opened by the "originating module" and the "terminating module"), application server 803 directs CCS 802 to transmit RTP streams (call content) directly between originating gateway 810 and terminating gateway 812, col. 15, line 24-46; the H.323 calls (i.e., terminating points) are not modified, the RTP streams are merely moved, col. 15, lines 45-46]**.

14. With regard to claim 16, Cave et al. discloses that the means for transmitting the call content out of the system is a media gateway **[Fig. 4c, terminating gateway 812]**.

15. With regard to claim 17, Cave et al. discloses that the originating half call context resides in a first media gateway **[Fig. 4c, originating gateway 810]** and the terminating half call context resides in a second media gateway **[Fig. 4c, terminating gateway 812]**.

16. With regard to claim 18, Cave et al. discloses that the originating half call context resides in a media gateway and the terminating half call context resides in the same media gateway **[this is interpreted as the disclosed special situation where calling someone in your local area**

requires an 800 calling card service (due to long distance variable costs plus local fixed costs access), col. 22, lines 12-18; thus, after CCS 802 learns that the IP address of the gateway that can reach the PSTN number is originating gateway 810's IP address (col. 14, lines 64-66), the terminating half of the call would go though the same gateway (e.g., moved-phone 832 would be attached to gateway 810)].

17. With regard to claim 19, Cave et al. discloses additional call contexts to allow for monitoring of the call **[Fig. 3, col. 13, lines 21-24; VRU can instruct originating gateway 610 to replicate packets and send them to the VUR (i.e., monitoring) in addition to gateway 626]**.

18. With regard to claim 20, Cave et al. discloses a method for processing calls in a voice over packet system, the system including a call controller having control modules **[Fig. 4c, VRU 800; call control server (CCS) 802, voice media server (VMS) 804, and application server 803]**, a media gateway **[Fig. 4c, originating gateway 810]**, an ingress channel **[Fig. 4c, PSTN connection between phone 814 and originating gateway 810]**, egress channel **[Fig. 4c, PSTN connection between terminating gateway 810 and moved-phone 832; this is interpreted as the disclosed special situation where calling someone in your local area requires an 800 calling card service (due to long distance variable costs plus local fixed costs access), col. 22, lines 12-18; thus, after CCS 802 learns that the IP address of the gateway that can reach the PSTN number is originating gateway 810's IP address (col. 14, lines 64-66), the terminating half of the call would go though the same gateway (e.g., moved-phone 832**

would be attached to gateway 810)] and a core packet network **[Fig. 4c, IP network 806]**, the method comprising:

receiving a call having call content, originating information, and terminating information on the ingress channel **[Fig. 4a, VRU 800 receives call (i.e., call content) from phone 814 (PSTN portion between phone 814 and originating gateway 810 is the ingress channel) through originating gateway 810 (caller ID identifies user—originating information), user inputs desired PSTN number of the desired party (terminating information), col. 14, lines 7-60; the call is interpreted as the call to the desired party—thus, only instantiated until after receiving the desired party's PSTN number]**;

establishing an originating half call context for the call based on the originating information **[Fig. 4b, PSTN network routes the call to originating gateway 810 (col. 14, lines 7-9); an RTP/RTCP session established between originating gateway 810 and VRU 800 in order to get the terminating information, col. 14, lines 33-34; Cave et al. also discloses using only control signaling between the gateways and the VRU and connecting the media stream to the destination (i.e., directly connecting the half calls between originating gateway 810 and phone 814 and between terminating gateway 810 and moved-phone 832), col. 21, lines 43-48]** the originating half call context having terminating points within the media gateway **[Fig. 4b, established at originating gateway 810, col. 14, lines 7-9; the H.323 calls (i.e., terminating points) are not modified, the RTP streams are merely moved, col. 15, lines 45-46]**;

controlling the originating half call context for the call by a first control module of the call controller **[Fig. 4, CCS 802 and application server 803 control the application for**

incoming requests for 800 calling card service (interpreted as an "originating module"), col. 14, lines 35-40];

instructing a second control module of the call controller to establish a terminating half call context for the call **[Fig. 4b, application server 803 then instructs CCS 802 to establish a connection between moved-phone 832 and VMU 800 via terminating gateway 810 (interpreted as a "terminating module"), col. 14, line 61 to col. 15, line 10]**;

establishing the terminating half call context for the call based on the terminating information **[Fig. 4b, a connection is established between moved-phone 832 and terminating gateway 810, col. 15, line 11-23]** the terminating half call context having terminating points in the media gateway **[Figs 4c, established at terminating gateway 810, col. 15, lines 13-14; the H.323 calls (i.e., terminating points) are not modified, the RTP streams are merely moved, col. 15, lines 45-46]**;

controlling the terminating half call context for the call by the second module **[Fig. 4c, application server 803 (i.e., using the "terminating module") monitors the call from terminating gateway 810 and moved-phone 832 and identifies the called party, col. 15, lines 15-20]**;

transmitting the call content from the originating context to the terminating context based on the controlling of each call context by the first and second control modules **[Fig. 4c, (while keeping open the connections opened by the "originating module" and the "terminating module"), application server 803 directs CCS 802 to transmit RTP streams (call content) directly between contexts in originating/terminating gateway 810, col. 15, line**

24-46; the H.323 calls (i.e., terminating points) are not modified, the RTP streams are merely moved, col. 15, lines 45-46]; and,

transmitting the call content out of the system on the egress channel **[Fig. 4c, PSTN connection between terminating gateway 812 and phone 832 is the egress channel]**.

Response to Arguments

19. Applicant's arguments with respect to claims 1, 4-6, and 9-20 have been considered but are moot in view of the new ground(s) of rejection.

Conclusion

20. The prior art made of record and not relied upon is considered pertinent to applicant's disclosure:

(a) Armbruster et al. (USP 5,892,811), Radio telecommunication network and method for unobtrusive call interception via transcoder bypass.

(b) Laiho et al. (USP 7,577,422), Lawful interception of multimedia calls.

(c) Picha et al. (USP 7,181,219), Wireless handover using anchor termination.

21. Any inquiry concerning this communication or earlier communications from the examiner should be directed to MARK A. MAIS whose telephone number is (571)272-3138. The examiner can normally be reached on 5am-4pm.

22. If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Pankaj Kumar can be reached on 571-272-3011. The fax phone number for the organization where this application or proceeding is assigned is 571-273-8300.

23. Information regarding the status of an application may be obtained from the Patent Application Information Retrieval (PAIR) system. Status information for published applications may be obtained from either Private PAIR or Public PAIR. Status information for unpublished applications is available through Private PAIR only. For more information about the PAIR system, see http://pair-direct.uspto.gov. Should you have questions on access to the Private PAIR system, contact the Electronic Business Center (EBC) at 866-217-9197 (toll-free). If you would like assistance from a USPTO Customer Service Representative or access to the automated information system, call 800-786-9199 (IN USA OR CANADA) or 571-272-1000.

November 19, 2009

/MARK A. MAIS/
Examiner, Art Unit 2467
/Pankaj Kumar/
Supervisory Patent Examiner, Art Unit 2467